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N92-14241

# A Minimalist Approach to Receiver Architecture<sup>1</sup>

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*This article describes new signal processing techniques for DSN radios and presents a proposed receiver architecture, as well as experimental results on this new receiver's analog front end. The receiver's design employs direct downconversion rather than high speed digitization, and it is just as suitable for use as a space-based probe relay receiver as it is for installation at a ground antenna. The advantages of having an inexpensive, shoe-box-size receiver, which could be carried around to antennas of opportunity, used for spacecraft testing or installed in the base of every antenna in a large array are the force behind this project.*

## I. Introduction

This article reports on research in progress into the design of a smaller, cheaper, and more reliable receiver for the Deep Space Network. Here, a receiver is defined as everything in the chain between the output of the first noise-floor-setting amplifier stage (the maser in the DSN) and the input to the telemetry detection and decoding equipment. The hardest link to forge in this chain is the receiver's analog front end, where the rf signal is downconverted and filtered before it is digitized. Therefore, this article concentrates on the basic signal processing scheme and analog hardware of the receiver, but not on the complex mixer to process the digitized outputs from the receiver's analog front end. The design presented exactly corrects amplitude and phase mismatch at one frequency, the carrier, or a calibration frequency; digital equalization can then eliminate phase and amplitude irregularities over the whole passband.

The proposed receiver architecture is a departure from the conventional approach to high-performance digital receiver design. Most designs push the point at which the signal is digitized closer and closer to the rf carrier frequency, grabbing each new advance in the speed of analog-to-digital conversion and digital signal-processing equipment, and exchange analog mixers and filters for their digital counterparts. Well-founded faith in the ability of digital electronics to process signals without degradation steers people in this direction. All such designs will, however, be limited by their sample-and-hold circuits. In fact, analog-to-digital converters are never really the limitation, since if sufficiently fast sample-and-hold circuits are available, they can simply be paralleled in order to capture higher bandwidth signals. For example, today's fastest digital oscilloscopes use four sample-and-hold amplifiers, which are fired in cascade. A sample and hold is very similar to a mixer, yet harder to build. Both operations can be implemented with a switch that opens and closes repetitively. However, the duty cycle of the switch in the mixer is 50 percent, whereas that in the sample and hold has to be made as short as possible.

<sup>1</sup> Work performed under a contract between JPL and Johns Hopkins University through the TDA Office.

The design outlined below replaces a high-speed sample and hold with a pair of mixers in a direct downconversion configuration. Preliminary results show that 80 dB of spurious-free dynamic range, far more than is necessary to cope with all except the very purest sources of interference, is achievable.

## II. Receivers

The task of a receiver is in effect to apply a matched filter to each symbol of the transmitted data stream. If the communications link is operating without coding, then the output of this filter will be a 1 or a 0 if the transmitter alphabet has only two letters and will represent one bit of information transmitted. For a larger transmitter alphabet, the output of the matched filter will be the index of the signal that was most likely sent. When the link uses coding, the receiver may benefit from not discarding so much of the analog information available about the signal. For example, for a binary signal set, the output of the matched filter should be a quantized version of the probability that a 1 was more likely sent than a 0, or something directly related to it. Binary phase-shift keying (BPSK) is one of the simplest forms of modulation used and is usually the choice for a communications system in which bandwidth is of little consequence, such as the link from a deep-space probe to the Earth. The signal structure and matched filter for BPSK in the presence of Gaussian noise appear in Fig. 1.

Each symbol is correlated with the transmitter's carrier. The result of this correlation is directly related to the probability that the symbol sent was a 1, e.g., if the voltage is positive then a 1 was more likely sent than a 0.

The block diagram of the matched filter in Fig. 1 is appealingly simple, and this article presents some experimental investigations into how closely a real receiver can approach Fig. 1's simplicity. Inasmuch as a balanced mixer is an analog multiplier, one might be tempted to try implementing the block diagram directly. The flaw with this scheme is the requirement that the mixer and subsequent very high-gain amplifier perform well at DC, where  $1/f$  noise causes difficulties. That is, many data patterns will place significant energy very close to the carrier. This dilemma can be solved by proper coding; however, the remaining problem of carrier tracking, i.e., producing a replica of the transmitter's carrier at the receiver, is extremely hard to solve, since the problems of shielding and component nonidealities are close to intractable.

The superheterodyne receiver shown in Fig. 2 is the traditional solution. This receiver isolates a piece of the ra-

dio frequency spectrum, which contains the desired signal and converts it down in frequency while also amplifying it. The staged downconversion and amplification eliminate the hazard of feedback and raise the signal level sufficiently so that, by the time its frequency gets close to DC,  $1/f$  amplitude noise in the mixer diodes and the following amplifiers is no longer important. Unfortunately, each stage of conversion is not only expensive, but also degrades the signal because the multiplications are imperfect. The sharpness of the available filters determines how many stages are necessary to drop from the rf carrier frequency to baseband. Thus, either higher  $Q$  filters or a higher digitizing rate are necessary to simplify the analog electronics in Fig. 2. The narrower an analog filter is, the harder it is to make it linear phase, i.e., to make it behave like a uniform delay for all frequencies in its passband. If this distortion were constant, it would not pose a problem since no information is lost, and, in theory, a digital equalizer could be used to correct the phase distortions of the analog filter. Unfortunately, high- $Q$  analog filters change with temperature and are not uniform over their passbands. Thus, even if the filter were in a thermostatically controlled oven, the equalizer would have to change every time the receiver was tuned to a different frequency.

## III. New Approaches

This section presents a receiver design in which only low- $Q$ , low-pass filters are needed, and most of the amplification takes place at a low enough frequency so that oscillation caused by feedback is not a problem. Only an extremely small amount of the incoming signal energy will appear close to DC. Even this loss could be eliminated by proper shaping of the transmitted signal. (For example, the pinned state convolutional codes described in [1], which the Galileo Big Viterbi Decoder is already able to decode, have this spectrum shaping capability.) The basic block diagram appears in Fig. 3.

The incoming signal is split in two and each half is mixed with one of two carriers which have the same frequency, but are as close to 90 deg out of phase as possible. The outputs from these two mixers are amplified, low-pass filtered, and digitized. The only filters necessary are the anti-aliasing filters for the analog-to-digital converters, and as each mixer sees only half the incoming power, dynamic range requirements are eased. If  $\beta = 0$ , then the two low-pass filtered signals are merely the two components of the Hadamard transform of the input signal and can be processed as shown in Fig. 4 to recover the spectrum one filter width to either side of the local oscillator.

This processing scheme is simply that of the image rejection mixer which has been around since the 1950s [2,3] and has been well studied. Researchers have constructed whole receivers based on this idea [4,5]. The most recent and successful design is in [6].<sup>2</sup> These designs, however, were all limited to fairly low frequencies (<70 MHz) and to applications where keeping costs low was more important than achieving the ultimate in performance. The crucial difficulty was that of making accurate quadrature power splitters—i.e., keeping  $\beta$ , the offset from perfect quadrature, very close to zero. These designs did not recognize the capability of a change in low-frequency signal processing to compensate for imperfections in the quadrature power splitter. When the quadrature power splitter is imperfect, all of the information in the original signal is still present, but the axes of the coordinate system in which the signal is viewed are no longer the usual orthogonal  $X$  and  $Y$ .

Figure 5 shows a simple geometric way of viewing the signal in its skewed reference frame. Both  $X$  and  $P$  are the outputs of the two receiver channels. For clarity, Fig. 5 shows the complement of  $\beta$ , called  $\alpha$ . Here 74 deg is greater than the worst skew that might occur in practice. Elementary trigonometry demonstrates that  $Y$ , the output of a channel in perfect quadrature, can be obtained easily as a linear combination of the two measured quantities  $X$  and  $P$ .

These straightforward results prove that proper baseband processing can eliminate the effects of small phase errors in the in-phase and quadrature local oscillators. Amplitude mismatches are of course even easier to deal with, as a simple rescaling is all that is required. Figure 6 shows the exceptionally simple baseband processing scheme necessary to eliminate both amplitude and phase mismatches, leaving a perfect in-phase and quadrature pair ready to feed into a complex mixer. Figure 3 envisions that all this processing will be done digitally; however, there is no reason why equivalent analog techniques would not work.

Unfortunately, the gain and phase mismatches will probably not remain constant over a reasonable operating temperature range and certainly not remain constant as the receiver is tuned. Thus, the constants in Fig. 6 really experience small variations with time scales on the order of a minute, and the remaining challenge is figuring out what these mismatches are in real time. One possibility is to switch periodically the input of the receiver

over to a calibration oscillator. This oscillator would have no stringent stability requirements since only the relative phases and amplitudes of the two channels are important. A more elegant solution, however, is to synchronously detect a continuously modulated tone. The amplitude of the tone will be low enough so that it can be subtracted out digitally without degrading the information-bearing signal, and the synchronous detector will ignore interference. Thus, calibration and reception can proceed simultaneously. Exactly the same technique can be used to calibrate the delay through the whole receiver system.

The two anti-aliasing filters ahead of the analog-to-digital converters will, of course, not be absolutely the same. These manufacturing imperfections will introduce, between the channels, a phase shift which varies over the filters' passband. Digital equalization can, however, compensate for these imperfections. The reason is that, just as with channel-to-channel phase mismatch, the phase shift does not cause any loss of information. Although digital equalization could, in principle, be used in any receiver, the direct downconversion architecture makes it completely practical, since the filters are low  $Q$  and thus stable with time and temperature. Also, the equalizer does not have to change as the receiver is tuned. One way to adjust this equalizer is to sweep the calibration tone.

#### IV. Hardware

Elementary trigonometry demonstrates the soundness of the correction shown in Fig. 5. However, these new ideas concern radio frequency circuits where poorly modeled effects can be very important. Thus, it is not obvious that machinery that implements the different block diagrams can in fact be built. Some sort of experimental verification is necessary. All sorts of mundane but thorny engineering complexities presented themselves in the construction of a prototype. The proposed design requires the low-noise, low-distortion amplification of signals ranging from audio to a few megahertz in the presence of rf local oscillator feed-through from the mixer, which is more than 100 dB stronger. Similarly, the diode quads used in the mixers must perform reasonably close to DC. The extremely rapid plunge in frequency has, however, some advantages that ease requirements on the mixers and amplifiers. Certain distortion products that plague multi-stage superheterodyne designs, especially those which use a stage of upconversion to improve image rejection, are completely absent in this direct downconversion approach. Most of the amplification takes place at sufficiently low frequencies so that substantial amounts of feedback can be used to reduce distortion. A great deal of effort has gone into the design of such amplifiers for transcontinental coaxial cables [7,8].

<sup>2</sup> Amplification of this reference by a personal communication from Polly Estabrook in the Communications Systems Research Section, Jet Propulsion Laboratory, Pasadena, California is acknowledged.

Construction of one of the two channels shown in Fig. 3 began in late 1990 in order to evaluate the following: the susceptibility of the mixers to second-order intermodulation products caused by radio frequency interference, the phase and amplitude stability of the mixers and filters, and the intricacies of the detailed design of the impedance-matching network and low-noise amplifier following the mixer. The design used a diode quad driven by a 1.8-volt local oscillator as the mixer. It was tested at both 100 and 500 MHz. These frequencies were chosen because of test equipment limitations, rather than because of any intrinsic circuit limitations. Without improvement, the device should work well above 1 GHz, but will still require an initial stage of downconversion in order to operate at X-band (8.5 GHz). Going directly from X-band to baseband would have its fascinations, but would also require that the receiver sit right on the back of the maser, as cabling losses are too great to transmit the signal any distance. If the low-noise amplifier has a temperature of 20 K, then 40 dB of amplification is necessary before the signal enters the mixer in order to override the attenuation of the mixer and the noise of the subsequent low-frequency amplifiers. With more effort, the noise figure at the mixer input could be reduced by 13 dB; the amount of amplification necessary would then drop by a similar amount. Such an improvement would be useful for a spacecraft command or probe relay receiver, but would not be of much use to the ground-based DSN.

The construction of the device presented many sticky problems in analog electrical engineering, though none was sufficiently interesting to report on. Figure 7 shows the disconcerting first view of the baseband spectrum. Power line pickup by the impedance-matching network following the mixer caused the spikes. Many iterations later, the same spectrum analyzer produced Fig. 8, which shows 80 dB of clear two-tone dynamic range. Two pure tones were fed into the mixer through a power combiner. The harmonics

were produced by the test sources themselves, not by the mixer or amplifier.

Figure 9 demonstrates the design's freedom from second-order intermodulation. The mass of tones on the right is a 90-percent AM-modulated carrier. The single peak on the left comes from a second signal generator, which was set 70 dB down in order to check for overload in the spectrum analyzer. Figure 10 shows the first of the dual-channel tests and demonstrates the amplitude tracking of the pair of low-pass filters. Their tracking relative to one another, the important criterion when digital equalization is used, is even better. The difference varied by less than 0.005 dB, the limit of measurement, while the temperature of the two filters cycled over a 10-deg C range. When a similar plot for phase is available, the performance of the complete receiver can be predicted with confidence.

## V. Conclusion

This article has presented a new way of digitally dealing with the imperfections of the analog hardware in a communications receiver, and has presented sufficient measurements to afford reasonable confidence that the overall architecture has no disastrous flaws. The design is unique in its incorporation of self calibration. The correction for unknown cable transmission delays between the receiver and antenna can also be handled by the same techniques used to measure the phase shift through the pair of low-pass filters. Cutting the number of analog filters in the design to the bare minimum and placing these at a point where the signal frequency does not change with receiver tuning gives the receiver great stability. All of the first-order (i.e., not time varying) imperfections in the two analog filters can be equalized out digitally, leaving only gradual second-order drifts to contend with. The great intrinsic stability of this receiver architecture makes it also suitable for use as a radio science receiver.

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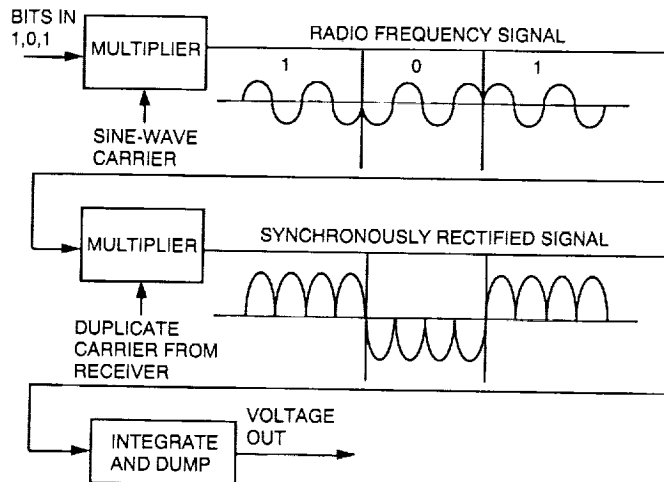


Fig. 1. BPSK detection.

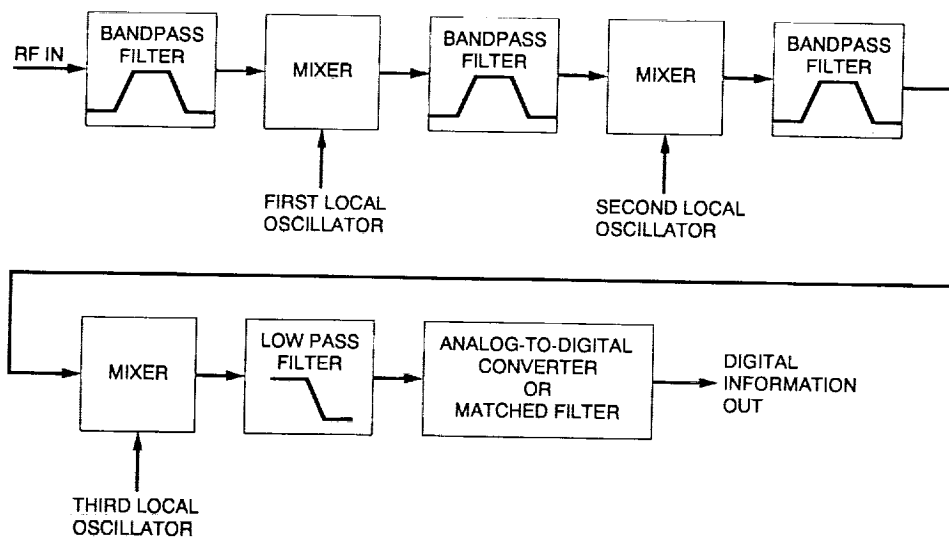


Fig. 2. Staged downconversion.

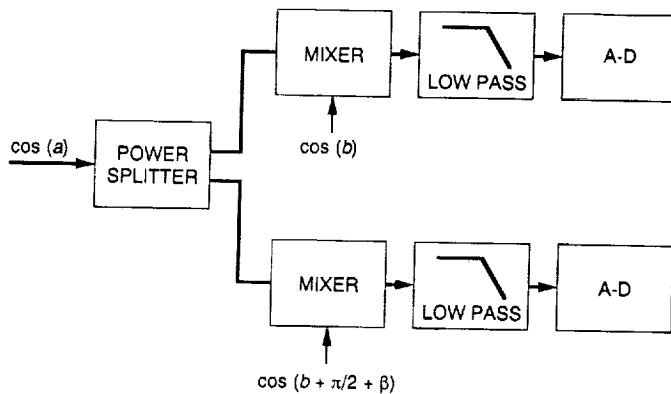


Fig. 3. Direct downconversion architecture.

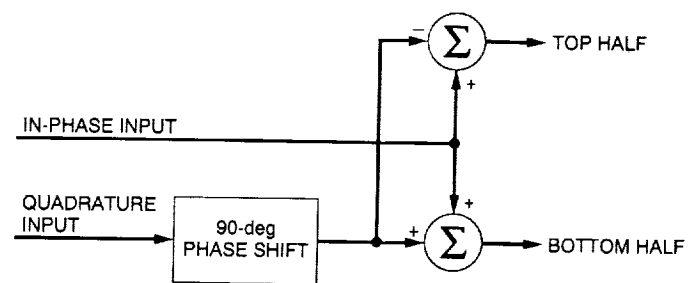


Fig. 4. Two image rejection mixers in one.

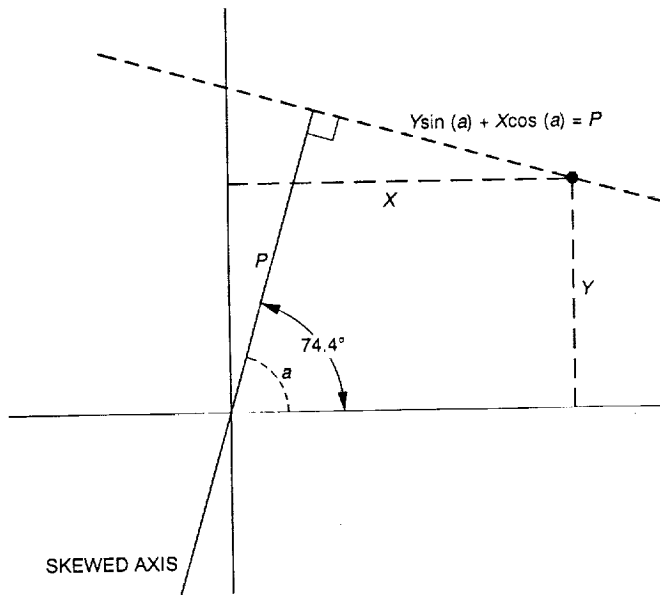


Fig. 5. Signal geometry.

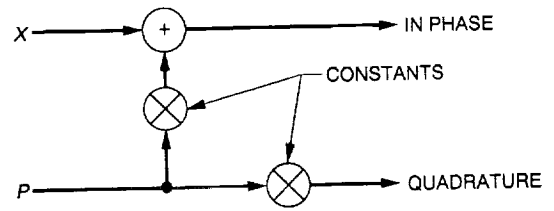


Fig. 6. Phase corrector.

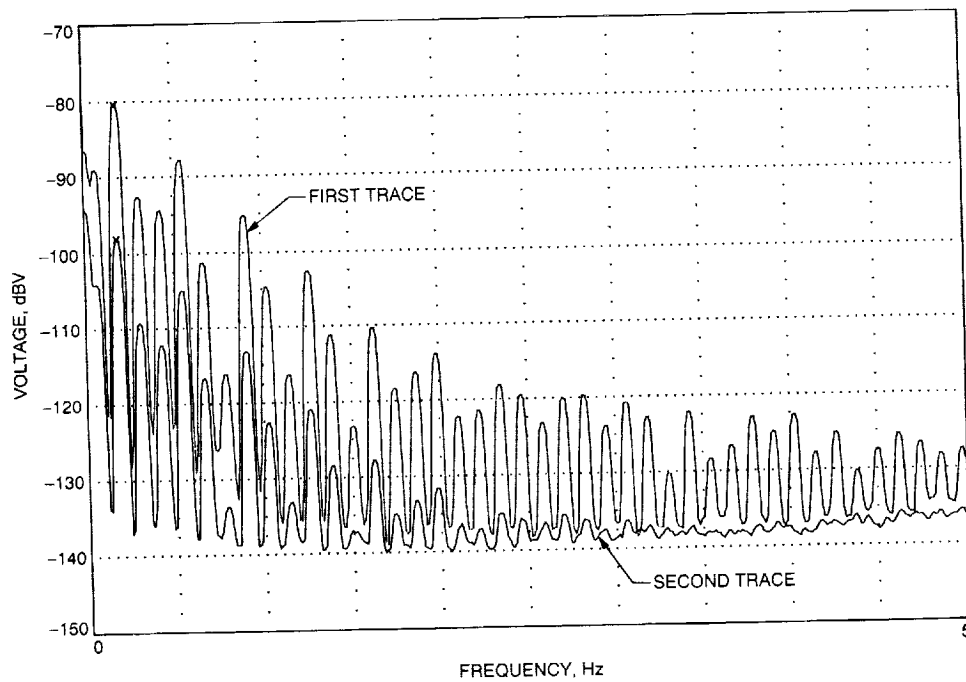


Fig. 7. Second-order distortion.

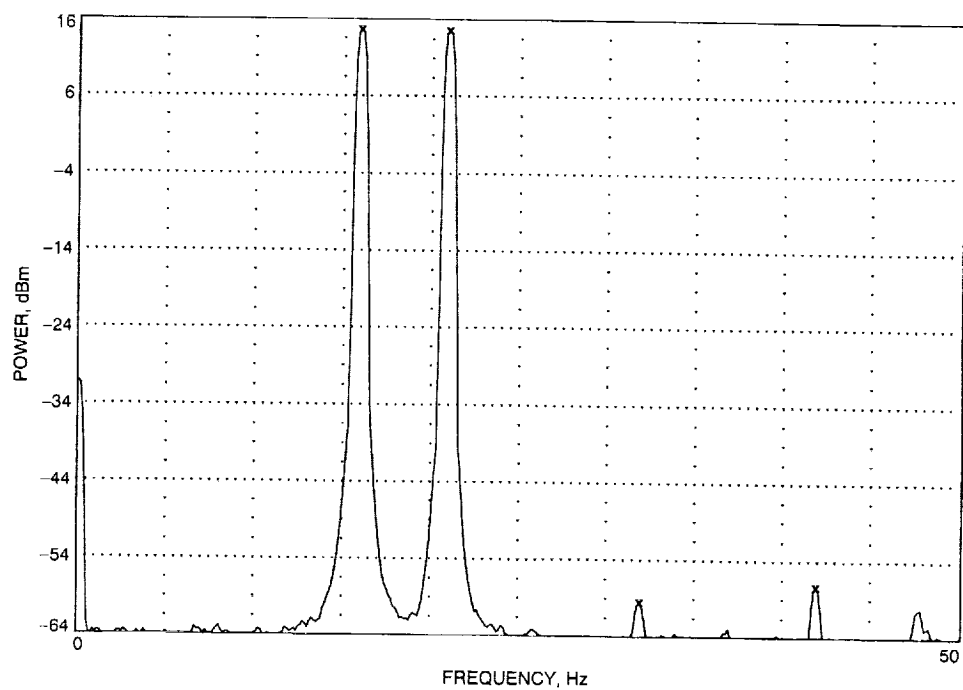


Fig. 8. Two-tone intermodulation distortion.

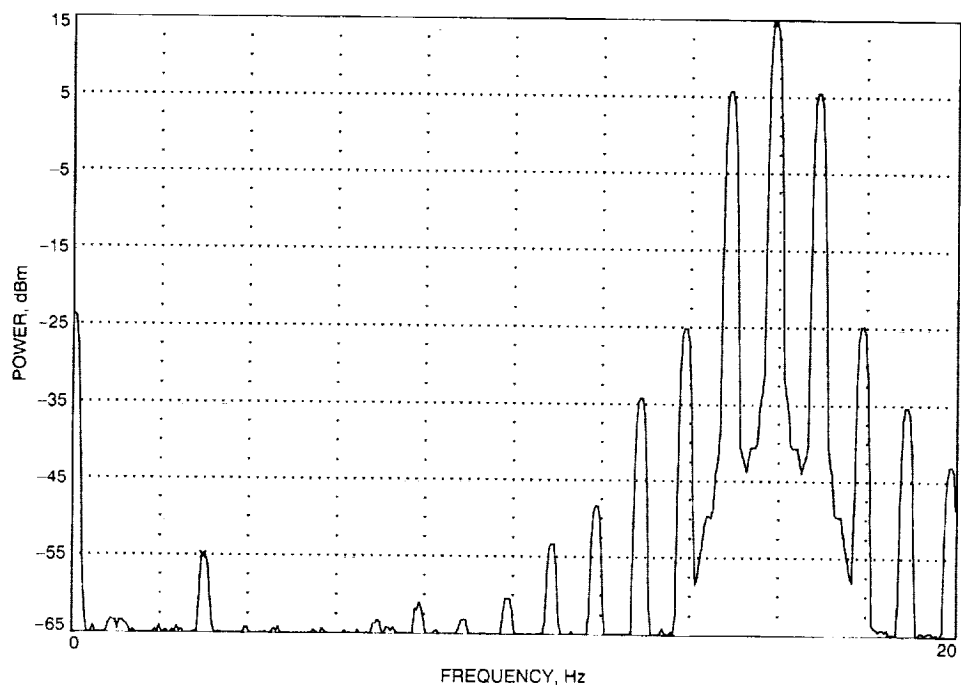


Fig. 9. Filter magnitude match.



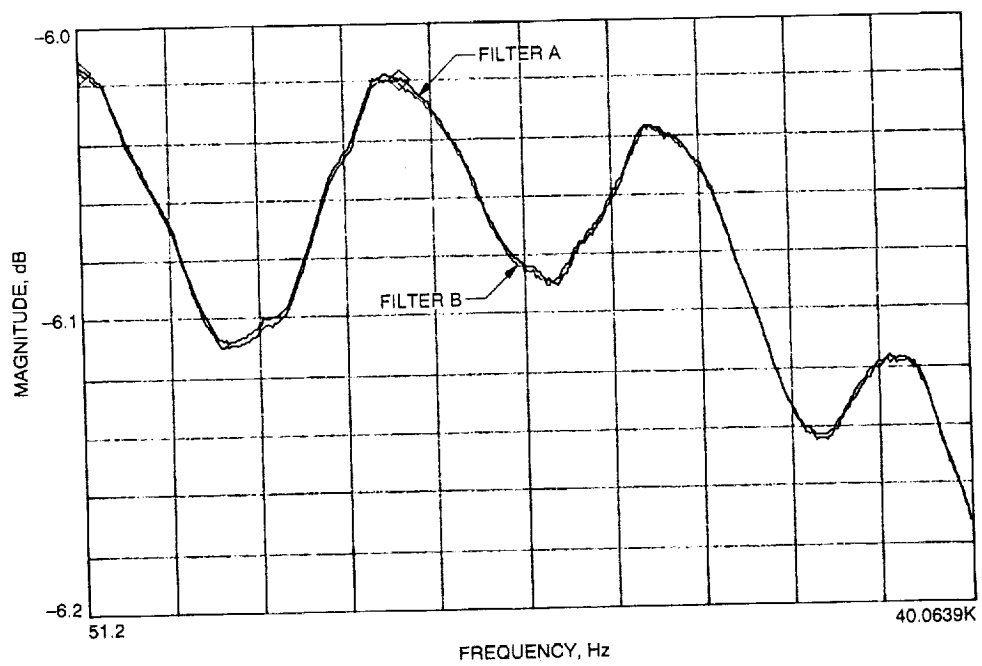


Fig. 10. Mixer output.